# Instantaneous Compandors on Narrow Band Speech Channels

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If speech is passed through an instantaneous compressor, the original speech frequency spectrum is substantially widened. It is known that instantaneously compressed speech can be transmitted over a medium with a passband no wider than that occupied by the uncompressed speech, and the original signals recovered without distortion. The conditions required for such distortionless transmission are examined. The analysis indicates that more severe requirements must be imposed on the attenuation and phase characteristics of the system when this reduced handwidth mode of operation is used. The practical value of this exchange of transmission requirements is a matter for experimental determination.

## Introduction

When a signal such as speech is instantaneously compressed in amplitude, harmonics and cross modulation products are generated which extend the frequency spectrum of the original signal by many octaves. It is proposed to demonstrate that the additional products thus generated are necessary for the distortionless recovery of the original signal. Then the conditions will be examined under which this broadband signal can be transmitted without distortion through a bandwidth no wider than that occupied by the spectrum of the uncompressed speech. Finally, some of the practical aspects of using intantaneous compandors on narrow band speech channels will be considered, with emphasis on the nature of the transmission requirements placed on the medium. The advantages to be obtained from the use of instantaneous compandors have already been presented by Mallinckrodt.[1]

## Bandwidth vs Distortion

If a single frequency tone is compressed by a 2 to 1 compressor,[2] and then the fundamental alone is expanded, it can be shown that the resultant 3rd harmonic distortion is only 13 db below the fundamental. Expansion of both the fundamental and the 3rd harmonic output of the compressor will reduce this 3rd harmonic distortion to 29 db below the fundamental. Expansion of the fundamental plus the 3rd and 5th harmonics will reduce the 3rd harmonic distortion in the recovered signal to 45 db below the funda-

[1] C. O. Mallinckrodt "Instantaneous Compandors," *B.S.T.J.*, Vol. XXX, No. 3, July 1951.

[2] In a 2 to 1 compressor, the output amplitude is the square root of the input amplitude. The name comes from the fact that, in such a compressor, the output amplitude will increase 1 db for each 2 db increase in input amplitude.

mental. These results are indicative of the significance of such components in the compressor output spectrum to distortion in the recovered signal.

## Frequency Analysis of Transmission under Reduced Bandwidth Conditions

It might be concluded from the results quoted above that a wideband channel is required for the distortionless transmission of instantaneously compressed speech. However, if the compressed speech is properly sampled

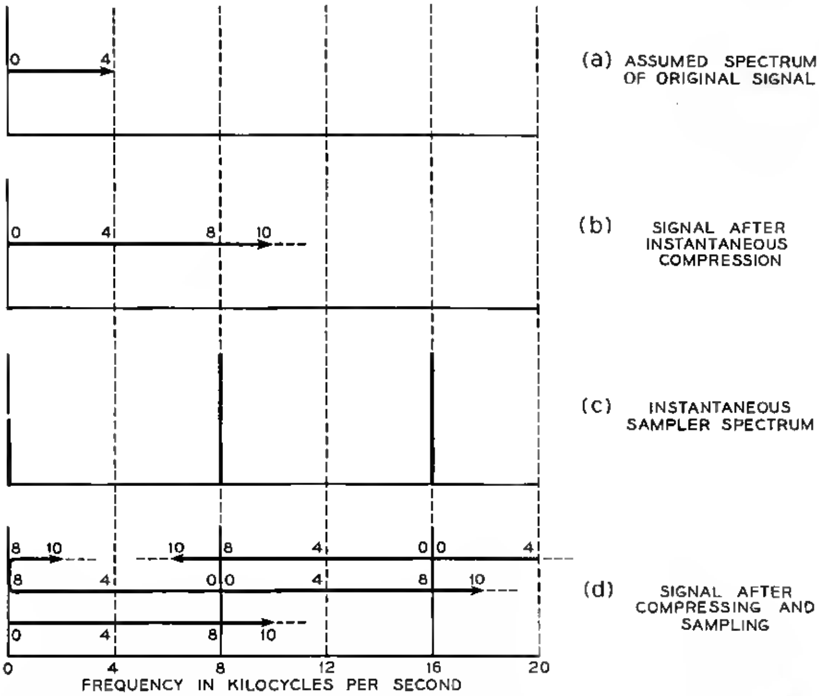


Fig. 1—Frequency analysis of instantaneous compressing and sampling of origina signal.

before transmission, and the received signals are again sampled at the receiver, the bandwidth of the intervening medium can be restricted to that of the original speech, and still the transmission can be distortionless. Hence, the sampling must transform the broadband spectrum of the compressed speech in such a way that it can be successfully transmitted over a relatively narrow band.

A steady state frequency analysis will serve to illustrate this phenomenon. Figure 1(a) shows the 4 kc frequency spectrum assumed for the original

signal, and Fig. 1(b) shows a 10 kc portion of this signal after instantaneous compression. Now the minimum sampling rate required to handle a 4 kc signal band is 8 kc. It is also the sampling rate that will allow tbe maximum band reduction in this case, as further analysis will show. The frequency spectrum of a sampling function with an 8 kc repetition rate has a d-c. component, an 8 kc fundamental, and all the harmonics of this repetition rate as shown in Fig. 1(c). These harmonics are all of equal amplitude and all are phased so as to add up every 125 microseconds to form the characteristic sampling waveform. Figure 1(d) shows the frequency spectrum formed by sampling the 10 kc portion of the compressed speech signal. It

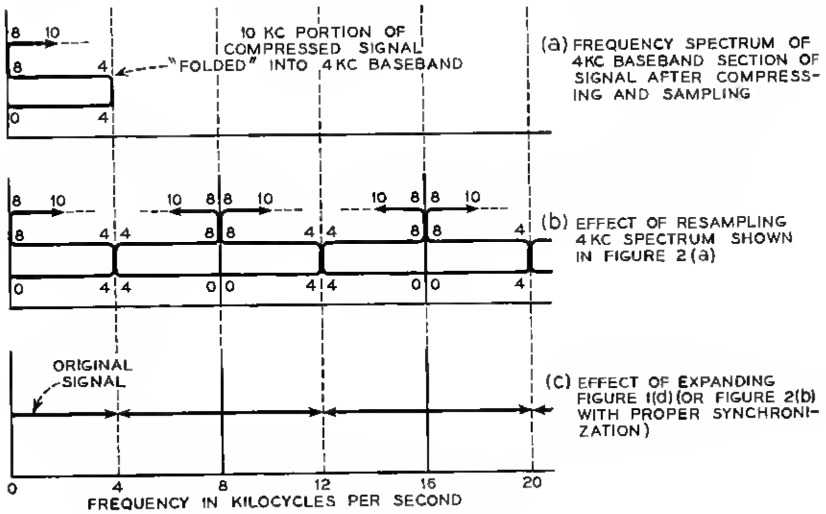


Fig. 2—Frequency analysis of instantaneous sampling and expanding of transmitted signal.

represents the product of the spectra of Fig. 1(b) and 1(c). As such, it is composed of the various component frequencies in the sampling spectrum as carrier frequencies, with the 10 kc portion of the compressed speech signals as amplitude modulated sidebands ahout these carriers.

Figure 2(a) shows the resulting spectrum that falls in the 4 kc haseband of Fig. 1(d). It represents that part of the compressed and sampled spectrum that would be received over an ideal 4 kc baseband channel. This spectrum is worth examining hecause it illustrates how the addition of instantaneous sampling makes it possible to transmit all the components in tbe compressed speech over a 4 kc channel. The effect may he described as a linear "folding" of the broadband spectrum back and forth over the 4 kc band. However,

althougb any broadband signal can be similarly folded into a 4 kc band by an instantaneous sampler with an 8 kc repetition rate, the process is not fully reversible. For example, there is no means of telling whether a 3 kc component in tbe folded signal comes from a 3 kc, a 5 kc, an 11 kc, or a 13 kc, etc. component in the original signal. Hence it is only a very special class of signals that can be recovered after their frequency spectra have been condensed in this fashion.

To recover the original speech in this case, the spectrum shown in Fig. 2(a) can be sampled at an 8 kc rate to produce the spectrum shown in Fig. 2(b). Now an examination of the spectra involved will show that when the second sampling is properly synchronized with the transmitting sampler, the two spectra shown in 1(d) and 2(b) will be identical. The spectrum in Fig. 1(d) represents the 8000 samples per second of the compressed speech generated at the transmitter. Thus, when the spectra of Figs. 1(d) and 2(b) are identical, samples will be recovered at the receiver which are identical to those that were generated at the transmitter. These can be converted to samples of the uncompressed speech by complementary instantaneous expansion. The spectrum of these samples is shown in Fig. 2(c). All that is necessary at this point to recover the original speech without distortion is to pass these samples through a 4 kc. low-pass filter.

## Requirements for Distortionless Transmission on Reduced Bandwidth Basis

Thus the criterion for distortionless transmission of compressed and sampled speech under these conditions is that the samples recovered at the receiver be the same as the samples of compressed speech that were generated at the transmitter. This means sending 8000 pulses per second over a 4 kc band without intersymbol interference. Nyquist[3] has shown that this is the maximum rate at which independent pulses can be transmitted over a 4 kc band and still be recovered at the receiver. At this maximum rate, the bandwidth employed does not give the transient response of one pulse time to die out before the next pulse is received. Therefore the transient response in this case must be such that, when one pulse is at its peak, the transient responses of all other pulses will be going through zero. The infinitely sharp cut-off at 4 kc, which is required to separate out the spectrum shown in Fig. 2(a) from that in Fig. 1(d), will have the required zeros in its pulse response, provided the attenuation is constant and the phase is linear with frequency.

This is the familiar $\frac{\text{Sin x}}{\text{x}}$ shape of transient response. Nyquist has shown

[3] H. Nyquist, "Certain Topics in Telegraph Transmission Theory," *A.I.E.E. Transactions*, Vol. 47, Pages 617 to 644, April 1928.

also that this is just one of a whole family of transmission characteristics with a specified symmetry about the cut-off frequency, all of which have the required transient zeros. However, there is no reason to suppose that any of them would prove less sensitive to variation of the transmission characteristics from the ideal than the one described above.

It is apparent that synchronization of the transmitting and receiving samplers is required to insure that the receiving sampling is done at the exact instant that all transient responses but the desired one are zero.

## Effect of Variations from Ideal Transmission Characteristics on Distortion

In practice of course a certain amount of distortion is tolerable. To get some measure of the practicability of such reduced bandwidth transmission of compressed speech, the first step is to determine how much intersymbol interference can be tolerated in this type of signal, and then to translate this tolerance into allowable variations in the frequency characteristics of the transmission medium from the assumed ideal. However, it is hard to estimate what the allowable intersymbol interference might be in this case. In a single channel system, intersymbol interference produces a form of distortion, and the sensitivity of such signals to distortion is primarily a matter for subjective determination.

For computational purposes it will be assumed, however, that this intersymbol interference should be 20 db down in the output. It is apparent that a 5% variation in the amplitude of a sample before expansion will produce a 10% variation in the expanded sample. On this basis 5% intersymbol interference in the medium between transmitter and receiver is the maximum allowable. Using Wheeler's theory of paired echoes[4], it can readily be shown that a sinusoidal variation in the phase vs. frequency characteristics of the medium, with an amplitude of $\frac{1}{10}$ of a radian (5.7 degrees), will cause a pair of echoes each of which will have a peak equal to 5% of the original sample. Similarly a sinusoidal deviation in the attenuation vs. frequency characteristic of 0.9 db from the ideal will also cause a pair of echoes with an amplitude of 5%.

In estimating the average effect of such echoes, it cannot be expected that the intersymbol interference from a given echo will be appreciably less than its peak amplitude would indicate. The principal reason is that, in order to realize the savings in bandwidth, the pulses are 125 microseconds apart, which is as close together as the 4 kc band will permit. Reference to the $\frac{\text{Sin } x}{x}$

[4] H. A. Wheeler, The Interpretation of Amplitude and Phase Distortion in Terms of Paired Echoes, *I.R.E.*, June 1939.

form of transient response indicates that the width of pulses (and hence of echoes) received over a 4 kc band, is such that they will be within 65% of their peak amplitude for a full 125 microseconds. Thus such echoes will cause at least 65% of their peak interference to at least one subsequent pulse. This illustrates why it is so difficult to control intersymbol interference in pulse systems operating under minimum bandwidth conditions.

Assuming from this argument that the interference from echoes should be taken at their peak values, the tolerable phase deviations from linearity must be measured in tenths of a radian in this case. On ordinary speech channels the tolerable phase deviations from linearity are measured in radians, which represents a difference of one or two orders of magnitude.

Another estimate of the allowable intersymbol interference may be obtained by comparing it to quantizing noise on a PCM system. A 5-digit PCM system has 32 quantizing levels, and the average uncertainty in the recovered pulse amplitude is one half of a quantum step, or approximately 1.6%. The 10% intersymbol interference requirement chosen above represents approximately 6 times as much deviation in recovered pulse amplitude. Again only subjective measurements can tell whether intersymbol interference in this case is six times more tolerable than quantizing noise. However, a 5-digit PCM system is not a high quality circuit by Bell System standards.

The distortion effects due to lack of synchronization of the transmitting and receiving samplers have been ignored in the discussion so far, on the assumption that it would not prove too difficult in practice to make it a relatively negligible source of intersymbol interference. However, it may not prove to be a negligible factor from the economic standpoint, when an attempt is made to prove in a system of this type.

## Multichannel Aspects

In the case of multichannel time division systems, the addition of instantaneous campandors seldom requires an increase in the transmission requirements of the medium. In multichannel PAM and PPM systems, for example, intersymbol interference causes intelligible crosstalk between channels, and the requirements on such crosstalk usually calls for the intersymbol interference to be some 60 db down in the recovered speech. In such cases the addition of an instantaneous compandor can serve to reduce this requirement on the line to some 40 db, through the so-called "Compandor Advantage"[5]. It is fair to point out, however, that such systems are seldom, if ever, operated as minimum band pulse systems.

[5] C. O. Mallinckrodt, "Instantaneous Compandors," *B.S.T.J.*, Vol. XXX, No. 3, July 1951.

## Conclusions

It has been shown that distortionless transmission of instantaneously compressed speech over a frequency band no wider than tbat required for the uncompressed speech does involve the transmission of a broad-band signal over a relatively narrow-band channel. This is made possible by the use of an instantaneous sampler which serves to "fold" the spectrum of the compressed speech at the transmitting end so that the entire spectrum is contained within the desired bandwidth. The criterion for distortionless transmission of these "folded" signals is shown to be one of recovering at the receiving end the precise samples of compressed speech that were generated at the transmitter. To accomplish this distortionless recovery of the transmitted pulses it is necessary, first, that the transmission medium cause no intersymbol interference, and, second, that tbe signals at the receiver must be sampled in synchronism with the sampling at the transmitter.

It was also shown that the full reduction in bandwidth can be realized only by pulse operation under minimum bandwidth conditions. It was estimated that the accuracy of control of the steady state phase and attenuation vs. frequency characteristics that would be required to maintain the intersymbol interference below an acceptable level would be bard to meet in practice, primarily because of having to operate under such minimum bandwidth conditions.